LEVEL II (12)

FINAL REPORT ON

HIGH FREQUENCY MODEM FOR ADVANCED NARROWBAND DIGITAL VOICE TERMINAL

Contract No. N00014-77-C-0580

TRW Report No. 32191.000

MARCH 1980

Prepared for
Naval Electronic Systems Command
Washington, DC 29360

TRW
DEFENSE AND SPACE SYSTEMS GROUP

ONE SPACE PARK - REDONDO BEACH, CALIFORNIA 90278

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HF MODEM FOR ANDVT

FINAL REPORT

1.0 INTRODUCTION

This report describes the Software developed for the HF MODEM for the ANDVT Contract. ~~Section 2.0 describes~~ the Transmitter and ~~Section 3.0~~ the Receiver are described. ~~Appendix 1~~ contains Flow Charts. Appendix 2, which is TRW Company Proprietary, contains the description of the Golay Encoder/Decoder, as well as the Flow Charts.

2.0 TRANSMITTER

2.1 Data Transmission Sequence

The following is the sequence of data transmission following Push-to-talk:

- NFP1 frames of Preamble Part 1
- NFP2 frames of Preamble Part 2
- 15 frames of PN
- 63 frames of MI
- 4 dummy frames
- SARK, VOICE of DATA transmission

2.2 Transmitter Status

The transmitter status can be determined at any time by monitoring the value of the TSF flag. This section describes the meaning of the different values of TSF and describes the activity of the transmitter in each state.

<u>TSF = 0 IDLE</u>

In the idle state the transmitter checks PTT and remains in the idle state as long as PTT is OFF. Note that TSF = 0 indicates that the Modem is a receiver. When PTT is sensed to be ON, the Modem becomes a transmitter and TSF is set to 1.

<u>TSF = 1 MI INPUT</u>

When TSF = 1, the transmitter is going through the steps required to input the MI and (if required) SARK data. The sequence of steps to get the MI can be described by discussing the values of TSFIF.

TSFIF = 0 DELAY

When TSFIF = 0, the transmitter delays GMIDLI frames and following the delay the MI REQUEST command is issued and TSFIF is set to 1.

TSFIF = 1 WAIT FOR MMI AVAILABLE

When TSFIF = 1, the transmitter checks MMI available. When MMI available is TRUE, TSFIF is set to 2.

TSFIF = 2 WAIT FOR FIRST PART OF MI

When TSFIF = 2, the transmitter checks the data in the input buffer and waits until the SYNC and the first part of the MI have been received. When the first part of the MI has been received, the 4 control bits are examined and one of the flags FLGVM, FLGDM or FLGSM is set to 1. The first part of the MI is moved from the input buffer and TSFIF is set to 3.

TSFIF = 3 INPUT SECOND PART OF MI

When TSFIF = 3, the second part of the MI is input. If FLGDM is 1, data mode initialization is performed and TSF is set to 2. If FLGVM is equal to 1, voice mode initialization is performed and TSF is set to 2. If FLGSM is equal to 1, then TSFIF is set to 4 to input the SARK mode data. If none of the 3 flags is equal to 1, then there is an error in acquiring the MMI and TSF is set to 12 for a TX shutdown.

TSFIF = 4 SARK DATA INPUT

When TSFIF = 4, the transmitter receives and stores away SARK messages until either SMDN messages are received or until PTT is OFF for SMDMPT frames. TSF is then set to 2.

TSF = 2 DELAY

This state is significant only in the DATA mode. The delay is needed to insure that the interleaver buffer has been filled (requires 174 frames) at the time it is necessary to send out the first DATA frame. There are NSKP2 frames of delay. NSKP2 = 1 for voice or SARK. TSF is set to 3 following the delay.

TSF = 3 PREAMBLE PART 1

When TSF = 3, preamble part 1 is transmitted for NFP1 frames. If in the DATA mode the interleaver buffer is being filled, Following NFP1 frames TSF is set to 4.

TSF = 4 PREAMBLE PART 2, PHASE REFERENCE

When TSF = 4, preamble part 2 is transmitted for NFP2 frames. If in the DATA mode the interleaver buffer is being filled. In the last frame during which TSF = 4, the 15 tone Phase Reference frame is transmitted and TSF is set to 5.

TSF = 5 PN TRANSMISSION

When TSF = 5, the PN sequence is transmitted. Following 15 frames TSF is set to 6.

TSF = 6 MI TRANSMISSION

When TSF = 6, the MI is transmitted. The MI transmission requires 63 frames. When the MI transmission is completed, then TSF is set to 7 for VOICE or DATA and TSF is set to 10 for SARK.

TSF = 7 4 FRAME DELAY

TSF = 7 indicates a 4 frame delay (needed for BCH decoding in the receiver).

In the DATA mode a dummy frame is transmitted and checks are made to determine whether the interleaver buffer was filled at the proper time. TSF is set to 8.

In the VOICE mode, the following is performed as SKP = 1, 2, 3, 4:

- If SKP = 1, the frame time is changed from 13.3 to 22.5 msec
- A dummy frame is transmitted
- If SKP = 1,2, or 3, then START is sent if SKP = STATFL and MOVEIN is called.
- If SKP = 4, MOVEIN is called and checks are made to see if single or double buffering of the voice input is required. TSF is set to 9.

TSF = 8 DATA MODE TRANSMISSION

If TSF = 8, the data mode executive DMEX is called to transmit a frame of data. If PTT goes OFF, then data is transmitted for approximately 184 frames and the transmitter goes to the IDLE state, TSF = 0.

TSF = 9 VOICE MODE TRANSMISSION

If TSF = 9, the voice mode executive VTEX is called to transmit a frame of voice. If PTT goes OFF, then approximately 10 frames are transmitted and the receiver goes to the IDLE state, TSF = 0.

TSF = 10 SARK MODE

When TSF = 10, the SARK data is transmitted. For each of the SARK mode blocks (SMDC contains the number of SARK mode blocks to be transmitted), 71 frames are transmitted. FC goes from 1 to 71. For FC = 1-8, the data is BCH encoded and dummy frames are transmitted. For FC = 9-71, BCH encoded data is transmitted. Following the transmission of the final SARK block, TSF is set to 11.

TSF = 11 DELAY WAITING FOR SHUTDOWN

TSF = 11 is used for a delay of NFPSMD frames following the last SARK transmission until shutdown. A dummy frame is transmitted each frame. TSF is set to 12 following the last dummy frame.

TSF = 12 TX SHUTDOWN

When TSF = 12, the modem returns to the 13.3 msec time (if necessary) and returns to IDLE.

2.3 Data Formatting

2.3.1 Preamble Part 1 - Doppler Tones

The data transmitted as Preamble Part 1 consists of 4 tones - 7, 13, 19 and 25 (and their conjugates - 57, 51, 45 and 39) transmitted continuously for NFP1 frames. The tone indices are based on a 64 point FFT numbered 0 - 63. Zero data is transmitted in all other tones.

2.3.2 Preamble Part 2 - Frame Sync Tones

The data transmitted as Preamble Part 2 consists of 3 tones - 10, 16 and 22 (and their conjugates - 54, 48 and 42) transmitted for NFP2 frames. The transmitted data is alternately multiplied by +1 and -1 each frame. The tone indices are based on a 64 point FFT numbered 0 - 63. Zero data is transmitted in all other tones.

2.3.3 PN

The PN sequence is 240 bits long and requires 15 frames. Sixteen dibits are formed from the 16 bits by bit replication; bit 0 becomes dibit 00 and bit 1 becomes dibit 11.

2.3.4 MMI

Each frame 16 consecutive dibits are selected.

2.3.5 Data Mode

Each frame 8 bits are taken from the interleaver buffer and are formed into 4 dibits. The 4 dibits are replicated 4 times to form 16 dibits.

2.3.6 Voice Mode

The data to be transmitted consists of the 54 input bits plus 24 additional bits generated by two Golay (24,12) encodings of selected bits. The bits selected for encoding are in tables TBLG1 and TBLG2. The 78 bits are formed into 39 dibits in accordance with tables VDMDB1 and VDMDB2.

2.3.7 SARK Mode

Each frame 16 consecutive dibits are selected as in the MMI.

2.4 Tone Assignment

2.4.1 16 Tone Mode

The 16 dibits are assigned to tones 8 - 23 (based on a 64 point FFT).

2.4.2 39-Tone Mode

The dibit/tone assignment is changed each frame as follows:

A table S(i) is formed by the rule

$S(i) = 5*(i-1) \bmod 39$; $S(1) = 0$, $S(2) = 5$, $S(3) = 10$, etc.

In frame 1, Dibit (i) is assigned to tone $S(i) + 12$

$i = 1,...39$.

In frame 2, Dibit (i) is assigned to tone $S(i+1) + 12$

$i = 1,...39$.

In frame n, Dibit (i) is assigned to tone $S(i+n-1) + 12$

S(i) is treated as a circular block in that the index of the block is reduced to mod 39 where needed.

The assignments are the same for frames 1, 40, 79, etc.

2.5 DPSK Modulation

The encoding table is as follows:

Odd Bit	Even Bit	Phase Shift
0	0	-225°
0	1	-135°
1	0	-315°
1	1	-45°

For the 16 tone mode, the modulation is on tones 8 - 23 based on the 64 point FFT. For the voice mode, the 39 tones are on tones 12 - 50 based on a 128 point FFT.

2.5 DPSK Modulation (Continued)

In order to avoid computation of sines and cosines in each frame, a table lookup technique is used as follows:

$\phi I(j)$ is th. initial phase of the tones. At any frame, the phase $\phi(j)$ will be of the form:

$$\phi(j) = \phi I(j) + k * 45° \text{ where } k = 0, 1, ..., 7.$$

The phase for any tone can be described by an integer between 0 and 7 and phase arithmetic is simply addition module 8. Four tables of sine and cosine are prestored as follows:

sine $(\phi I(j))$, $\cos(\phi I(j))$, sine $(\phi I(j)+45°)$ and $\cos(\phi I(j)+45°)$

The sine and cosine of $\phi(j)$ are just the values in the jth position of the 4 tables (to within a sign change).

Note that in the PN mode, the only dibits are 00(-225°) or 11(-45°). This results in the modulation being a biphase DPSK modulation.

3.0 RECEIVER

The detailed logic of the receiver is described in the flow chart and narrative of subroutine RXEXEC which can be found in the flow chart section. Here we describe the receiver operation more generally. The following sections contain a description of the main receiver computational segments as follows:

- PRELIMINARY COMPUTATION
- SIGNAL PRESENCE COMPUTATION
- FRAME SYNC COMPUTATION
- DOPPLER TRACKING
- DEMODULATION
- PN LOGIC
- MMI LOGIC-BCH DECODING
- GOLAY DECODING
- DATA MODE DEINTERLEAVING

3.1 Preliminary Computation

By "preliminary computation" we mean the processing starting with the moving of raw data from the A/D input buffer and ending with the computation of the absolute value of the FFT output. This computation is performed by subroutine RPREP and consists of calls to a series of subroutines. We will describe the operation of RPREP by describing each of the subroutines.

In the description which follows, NW = the number of samples per frame. NW = 162 for a 22.5 msec frame and NW = 96 for a 13.3 msec frame.

3.1.1 Raw Data Buffering (DMOVE, CMOVE)

These subroutines maintain a buffer of length (NW+60) which contains raw A/D data. Subroutine DMOVE moves the last 60+NS words from the end of the buffer to the start of the buffer. Subroutine CMOVE moves NW+NS words from the A/D input buffer to the remainder of the (NW+60) word buffer. NS is the frame sync offset computed by the Frame Sync program. This type of overlap is needed to provide for a continuous block of data and to provide for data in the "head" and "tail" for the Hilbert transform convolution.

3.1.2 Hilbert Transform (HILBRT)

The Hilbert transform is performed by convolving the NW+60 point input buffer with the following convolution operator:

3.1.2 Hilbert Transform (Continued)

.006, 0, .0122, 0, .0261, 0.0505, 0, .0908, 0, .1605, 0, .3102, 0, 1, 0, -1, 0, -.3102, 0, -.1605, 0, -.0980, 0, -.0505, 0, -.0261, 0, -.0122, 0, -.006.

The convolution is performed at NW+10 points of the NW+60 point input buffer and produces a (NW+10) point output in buffer HILOUT. The Hilbert convolution results in a "gain" of 1.585 which must be compensated for in Doppler correction.

3.1.3 Data Buffering and Doppler Correction (FNLMVE, DOPCCR)

In order to do the frame sync computation, Doppler corrected data for more than 2 frames must be available. For ease of indexing, data for 3 full frames is maintained in a block COROUT of length 3*NW. Subroutine FNLMVE moves the last 2*NW-NS words from the end of buffer COROUT to the start of the buffer. Subroutine DOPCOR performs the Doppler correction and fills the last NW+NS words of COROUT. The Doppler correction involves a complex multiply in which only the real part is saved. The input point is considered as the real portion and its Hilbert transform is considered as the imaginary portion. The "complex poin " is multiplied by a Doppler correction complex value (computed by subroutine VCOSTP) and the real part of the result is stored in the buffer COROUT.

3.1.4 FFT Computation (FFT, SQRFF, FFTUP)

For the 22.5 msec frame, a 128 point FFT is performed using as the input the middle 128 points of the 486 length buffer COROUT.

For the 13.3 msec frame, a 64 point FFT is performed using as the input the middle 64 points of the 288 length buffer COROUT. Subroutine SQRFF is called to compute the absolute values and subroutine FFTUP is called to scale the data up by FFTNUP places.

3.2 Signal Presence Computation

3.2.1 Signal Presence Check - Preamble Part 1, Part 2

The input to this check consists of the absolute values of the output from the FFT. Only frequencies 0-32 (of the 64 point FFT) need be considered due to the FFT symmetry.

The signal presence check for the preamble consists of comparing the energy in the designated tones with the energy in the other tones. For this test we define signal (S) as the sum of the energy in the

3.2.1 Signal Presence Check - Preamble Part 1, Part 2 (Continued)

designated tones plus the tone one higher and one lower than the designated tones. For preamble part 1, we would define S as the sum of the energy in the 12 tones 6-8, 12-14, 18-20 and 24-26. We define noise (N) to be the sum of the energy in the other tones; tone 0 and tone 32 are not used.

The signal is assumed to be present if

$$S - K2 * N > 0$$

We are using K2 = .68 for preamble part 1 and K2 = .72 for preamble part 2 (SPCPR1 and SPCPR2 are the labels).

3.2.2 Signal Presence Check - 16 Tone, 39 Tone Mode

The input to this check consists of the absolute values of the output from the FFT. The check consists of comparing the energy in the tones data is expected (S) with the energy in the other tones (N).

The signal is assumed to be present if

$$K*S - N \geq 0$$

K = 24/(39*2.7) = .23 for the 39 tone mode

K = 15/(16*2.52) = .37 for the 16 tone mode

3.3 Frame Sync Computation

The frame sync computation is done via a DFT followed by a filter. The DFT is performed on the Doppler corrected raw input data and overlaps 3 frames. The input data to the frame sync computation consists of 3 consecutive frames of NW points. The data is numbered 1 - NW for the previous frame, NW+1 - 2*NW for the current frame and (2*NW+1)- 3*NW for the next frame.

3.3.1 Frame Sync Computation - Preamble Part 2

Three sets of overlapping DFT's are performed (one at each of the three preamble part 2 frequencies). The DFT's are 96 points. The "early gate" DFT is over points 73 - 168 and the "late gate" DFT is over the points 119 - 204. If we define E_i, L_i as the absolute value of the DFT results for the ith frequency, then the input to the filter is $DF = [(E_1-L_1) + (E_2-L_2) + (E_3-L_3)]/[E_1+L_1) + (E_2+L_2) + (E_3+L_3)]$

3.3.1 Frame Sync Computation - Preamble Part 2 (Continued)

The "filter" consists of multiplying DF by a constant PR2KC = -11. The frame sync offset, in units of "input points" is thus PR2KC * DF. The sign indicates early or late.

3.3.2 Frame Sync Computation - 39 Tone Mode

A single overlapping 128 point DFT at the 11th frequency is performed. The "early gate" DFT is over points 148 - 275 and the "late gate" DFT is over points 210 - 337. If we define E and L as the absolute values of the early and late gates, then the value passed to the filter is DF = (E-L)/(E+L).

3.3.3 Frame Sync Computation - 16 Tone Mode

A single overlapping 64 point DFT at the 7th frequency is performed. The "early gate" DFT is over points 89 - 152 and the "late gate" DFT is over points 135 - 198. If we define E and L as the absolute values of the early and late gates, then the value passed to the filter is DF = (E-L)/(E+L).

3.4 Doppler Tracking

3.4.1 Doppler Tracking - Preamble Part 1

The Doppler Tracking program examines the power in the Doppler tones and the tones to each side of the Doppler tones. The input to the Doppler tracking is the absolute value of the FFT output. In the preamble part 1 Doppler tracking, the Doppler tones are 7, 13, 19 and 25. The computation performed is as follows:

Define E(j) as the energy in tone j.

$$XMH = E(7) + E(8) + E(9) + E(12) + E(13) + E(14) + E(18) + E(19) + E(20) + E(24) + E(25) + E(26)$$

$$EOUT = E(9) - E(7) + E(14) - E(12) + E(20) - E(18) + E(26) - E(24)$$

FIN is the input to filter and is defined as

FIN = 16 x EOUT/XMH

Prior to passing FIN to the filter, FIN is hard limited at .375 or -375.

The filter performs the following computation:

F1 = 128 x FIN

F1 is hard limited at 400 in the preamble mode

3.4.1 Doppler Tracking - Preamble Part 1 (Continued)

The filter output is FOUT where

FOUT = FOUT + F1.

FOUT is passed to the VCO subroutine which multiplies FOUT by a constant to convert from filter units to Doppler correction in degrees.

3.4.2 Doppler Tracking - 16 or 39 Tone Mode

In the 16 or 39 tone mode, Doppler Tracking is derived by measuring the demodulation phase error. The weighted phase error (weighted by the energy in the tone) is computed each frame and filtered. A more detailed description is found in the flow charts.

3.5 Demodulation

The demodulation function is performed in subroutine VBITCM. The input to the demodulation program consists of the FFT outputs (real and imaginary) for the tones for the current frame and for the previous frame.

The subroutine performs the following:

- The previous frames output is adjusted to compensate for phase differences introduced because of frame sync induced shifts.
- The computations

  S1 = Im (k) * Re (k-1) - Re (k) * Im (k-1)

  S2 = Re (k) * Re (k-1) + Im (k) * Im (k-1)

  are performed. k refers to the current frame and (k-1) to the previous.
- When necessary, S1 and S2 are added to the proper di[illegible]sity summing registers. The logic to go from S1 and S2 to the dibit is performed after all diversities are summed. The dibits are decoded by the following rule:

S1.GE.0	B1 = 1
S1.LT.0	B1 = 0
S2.GE.0	B2 = 0
S2.LT.0	B2 = 1

- The assignments of tone to dibit and dibit to bit are performed.

3.6 PN Logic

Subroutine PNEX, the PN executive, is executed each frame until a PN match is found. The program maintains a string of 240 bits; stored as fifteen 16 bit words. Each frame, 16 bits are computed. The most recent 240 bits are compared to the 240 bit PN pattern. If the number of bit matches is greater than PNTH (nominally 60), a PN match is assumed and the MMI processing is started.

3.7 MMI Logic-BCH Decoding

The BCH Decoding requires 67 frames. The details of the BCH decoding are in the flow charts. We will describe here what takes place in each of the frames:

Frames 1 - 55	-	Demodulation and diversity combining
Frame 56	-	Demodulation and diversity combining, computation of bits 1-28 and partial syndrome computation using those bits (3 bits all equal to 0 are artificially introduced)
Frame 57	-	Demodulation and diversity combining for bits 29-60; computation of bits 29-60 and partial syndrome computation using those bits.
Frame 58	-	Same as above for bits 61-92
Frame 59	-	Same as above for bits 93-124
Frame 60	-	Same as above for bits 125-156
Frame 61	-	Same as above for bits 157-188
Frame 62	-	Same as above for bits 189-220
Frame 63	-	Same as above for bits 221-252

At this point we have 255 bits having introduced 3 zero bits.

Frame 64	-	Berlekamp iterative algorithm (if needed) and Chien error search bits 1-7; these include the control bits
Frame 65	-	Chien error search bits 8-52 (if needed)
Frame 66	-	Chien error search bits 53-97 (if needed)
Frame 67	-	Chien error search bits 98-131 (if needed)

3.8 Data Mode Deinterleaving

Each data mode frame 8 bits are computed and placed "vertically" in a buffer which is 58 by 24 bits. While this buffer is being filled, a second buffer is being emptied "horizontally". Each 3 frames 24 bits are taken from the buffer and a "hard" Golay (24,12) decoding is performed. The decoded 12 bits are output 4 bits at a time for 3 frames. Every 174 frames the buffers are switched.

3.9 SARK Mode

In the SARK mode, the receiver performs BCH decoding as in the MMI. There is a 71 frame "period" of operation. During frames 1-67, the receiver performs the same functions as in the MMI. Frames 68-70 are dummy frames. In frame 71, the BCH decoded SARK data is packed and stored and initialization for the next SARK message is performed.

3.10 VOICE Mode

In the voice mode, 78 bits are demodulated. Two sets of 24 bits are extracted from the 78 bits and a "soft" Golay (24,12) decoding is performed for each set. The decoded data bits are saved resulting in 54 bits. The bits are packed into 9 words (6 bits per word) and output.

3.11 Golay Decoder

The Golay decoder is included as a TRW Proprietary addendum.